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ATM versus IP for Voice over IP

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Abstract: Being packet or cell switching technologies, both IP (Internet Protocol) and ATM (Asynchronous Transfer Mode) have been competitor of each other for a long time based on business needs, features, functions and capabilities. Suitability of ATM and IP for carrying voice and the advantages of one over the other has been a contradictory issue. On one hand IP (Internet Protocol) has the lead of being preferred choice due to lower cost, higher adoption rate and advanced features. On other hand ATM provides many advantages including higher quality of service, especially for time-sensitive materials because of its bandwidth efficiency. There are advocates of both of these technologies which claim that each is suited to particular needs and environments and not complementary of other. QoS (Quality of Service) is the main deciding factor for choosing best of these both technologies that suit the requirement of end-to-end traffic delivery. This paper shows a comparison of performance between ATM and IP for Voice over IP using OPNET simulation model. Obtained results are compared and discussed in details.

Keywords: Voice over IP, ATM, QoS, OPNET, Simulation

I. INTRODUCTION

Quality of Service (QoS) in context of data networks is known to be the ability of network to guarantee that networks is capable to give better results.

The aim behind the development of ATM networks was to integrate services such as voice and video traffic into a single network. In contrast, IP networks were developed primarily for computer data communication. Generally, computer data communication is more efficiently supported by a connectionless network service.

There has been a difference of opinion between those who believe that ATM networks are the way forward and those who believe that with some modification IP networks can provide the best solution. Some researched believe and advocate that the exclusive use of ATM, even for connecting devices within the end system (Hayter. *et al.*, 2001). Conversely, some claim that IP networks can provide any of the services that ATM can support. Thus the performance of both these technologies has been compared from different angles and contextual settings. This research paper is focused on indenting and comparing the performance of both ATM and IP technology with the help of some simulations.

2. ATM

Asynchronous Transfer Mode uses only small fixed size packets known as cells. It performs simple functions in the transit nodes, with a header containing simple error detection, error checking and recovery of the cell contents. Its connection oriented information transfer is allow fast and simple hardware switching in the network, no matter which type of traffic it forwards. The low bandwidth traffic will use the cells which contain less bandwidth than high bandwidth traffic for example video, in which use of cells is not synchronized.

A. *ATM Protocol Stack*

The main concept of the ATM layer is independent of the physical mechanism which is used for transmission, it sends data passing towards down by ATM adaptation layer to its destination. The transmission convergence sub-layer allows an interface to the transport medium, where two important functions are cell delineation and header error control generation. However, ATM is unable to provide functions like retransmission of corrupted data. ATM Adaptation layer (AAL) indicates end-to-end protocol that gives interface between ATM layer and higher layer protocols with the applications. The main function of the AAL is to accept messages from higher level protocols and dividing them into smaller entities for the purpose of transport into the cells.

B. *ATM Services*

The Main function of the ATM is to support four traffic types that are CBR, VBR, ABR and UBR:

(1)CBR: Constant Bit Rate is designed for the purpose of carrying synchronous time sensitive traffic for example as voice.

(2) VBR: Variable Bit Rate contains two sub-classes which are real time(RT) or non-real time, VBR is for that services in which the data stream is not supposed to be transported at a constant rate, for example as video. (Frossard, *et al.*, 2001)

(3) ABR: Available Bit Rate is designed for those applications where there is no issue of delay, when the

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capacity is more the ABR sources move at a faster rate, but when the load of VBR increased then ABR sources are decreased their rates. (Chao 2000)

(4) UBR: Unspecified Bit Rate indicates as a low cost low QOS service that gives no performance guarantee. (Breivik, *et al.,* 2002).

C. *ATM Adaptation Layer (AAL)*

It consists of four basic services, each service is to support many applications over an ATM network and the function of each service is to generate in the form of switching mode, bit rates and delay tolerance.

D. *ATM Switching*

The main concept behind ATM is high speed hardware based switching. ATM switches are normally multiplexors, which deal with the situation such as, when two cells arrive at same time they are required to leave from the same port and it can be resolved by buffering. However both affect QOS by adding delay, which can be controlled later by increasing the cell loss rate.

3. INTERNET PROTOCOL (IP)

IP is the fundamental pillar of Internet. The Internet protocol was formed for internetworking as a normal protocol almost to carry any network. IP allows a 'best effort' service over network layer as a datagram service, data from the transport layer (TCP or UDP) is changed into IP datagrams and moved over the network. IP network is a network made up of nodes. Internet protocol version 4 (IPV4) standard which gives the knowledge of IP addresses as four bytes (i.e,32 bits) in length and the advanced Internet protocol version 6 (IPV6) standard that shows the addresses as sixteen bytes (which is 128 bits) in length. Internet protocol network contains a data which is created into the packets and every IP packet consist of header (which indicates the source, destination and the details of data) as well as the message data itself.

A. *QoS or IP*

The Transmission Control Protocol/Internet Protocol (TCP/IP) stack provides a 'best effort' service. However different applications need of several applications that must be more tightly constraints during the transfer of information. Many mechanisms established for the purpose to allow QoS aspect to transmission of IP.

B. *Integrated Services*

The aim of integrated services is to specify that the elements of network guarantee to deliver performance based on quality of service (QoS). The idea behind integrated services suggests that each router in the network support every application to be given due priority.

C. *Resource Reservation Protocol (RSVP)*

The essential point of the integrated services and QOS which are related to internet is Resource Reservation protocol. RSVP is receiver-oriented as it request for the resources which can be reserved between destination and source. RSVP arrange this by forwarding special IP packets as a PATH message which intimates the routers between the source and destination to manage the 'reverse path' when the receiver receives the PATH message. The RESV message includes the QoS specifications for receiver as well as for transmitter.

Once the reservation takes place, the router knows that it has full resources to connect the QoS requirements, it transfer the RESV message to other router in the form of tree. This procedure keep going on to the RESV message meet to source. In RSVP, RESV message are transferred by the interval of 30 seconds to maintain the connection. If router does not get any update it deletes the path. (Chen, *et al.,* 2000).

4. VOICE OVER INTERNET PROTOCOL

VOIP technology provides telephone calls over the internet using Internet Protocol (IP). It mange to send analogue voice signals into data packets. VOIP depends on H.323 standard to save cost for making long distance telephone calls. The only disadvantage of VOIP is call drop ratio and low quality of voice. (White, *et al.,* 1998).

The aim of VOIP is promote voice communications over packet switched networks. The VOIP packet switched networks can be divided into fixed wire Local Area Network (LAN), fixed Wide Area Network (WAN) and wireless links. Implementation of VoIP indicates standardised solutions that are the International Engineering Task Force (IETF), and International Telecommunications Union (ITU). However, speech codes are normally used during VoIP calls. These are often selected from the ITU group of codecs as G.711, G723.1 and G729 and G729.1. Each of them facilitates VOIP to provide better quality as compared to the telephone calls which are standard. **(Fig .1)**

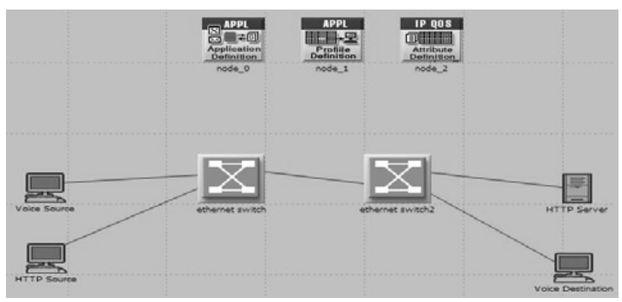

Fig 1: Using IP topology for VoIP

A. *Session Initiation Protocol*

The SIP is the most famous protocol for voice over internet protocol. SIP is the text based protocol with simple implementation and it consist several syntax-related equally to the Hyper-Text Transfer Protocol (HTTP) basically usage in the internet. A SIP call consist of many different network elements and other as clients, servers, proxies and registrars.

A client indicates the end point of SIP call and is known as network element which transfer SIP requests and receives SIP responses where human user contact with a client for the purpose to set up and receive VOIP calls.

A SIP server is the network element that gives the process of SIP requests and returns back the responses, SIP proxy is the network element that includes server as well as client and it will pass messages with in the two SIP clients, and the last is registrar at that will take over registration request from the SIP client and therefore it is instrumented for location of SIP clients by allowing a located service to the servers of SIP. **(Fig.1to14)** (Dang, *et al.*, 2004).

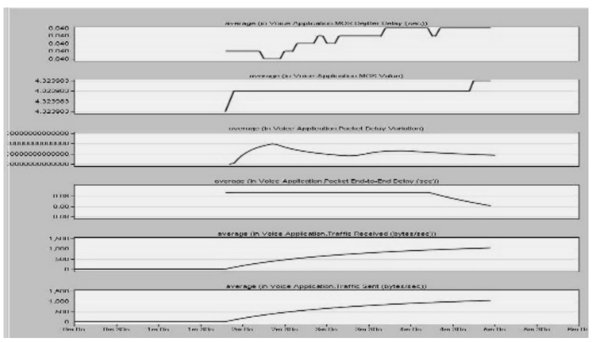

Fig. 2: Voice source simulations for voice

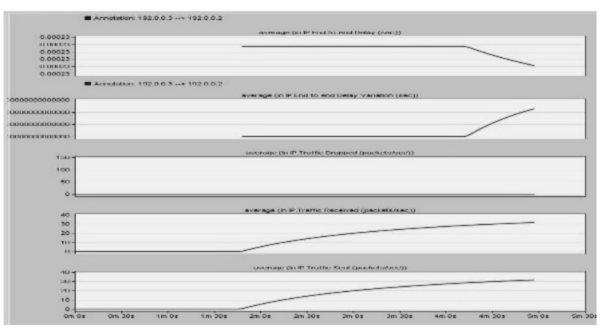

Fig.3: Voice source simulation for IP

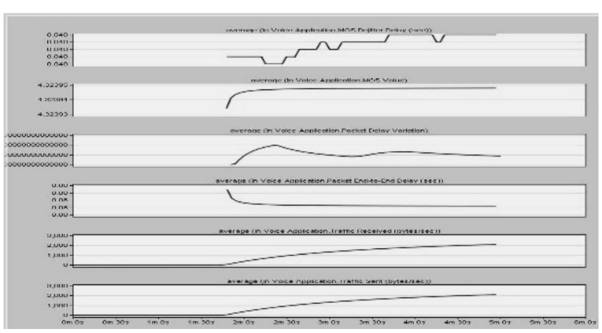

Fig. 4: Simulation for response of HTTP with voice

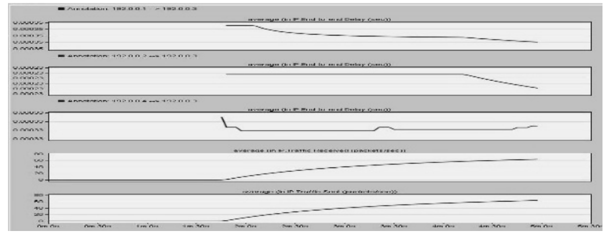

Fig.5: Simulation for response of HTTP over IP

B. *Prioritization*

Prioritization is linked with the QoS, the IP QoS protocol is RSVP, which gives the sender for the purpose of requesting a collection of handling traffic, nowadays the group of IETF which is the Insert working group establishing a solution, the model which consist of Differentiating Services used the Type of Service (ToS) octet field of IP header to distribute traffic in between the service provider and the customer.

C. *Voice Compression*

Compression plays a main role in VOIP because of slow speed links for the flow of traffic as the enterprises which are medium size at the speed of 28.8 kbps are linked to the virtual private network (VPN).

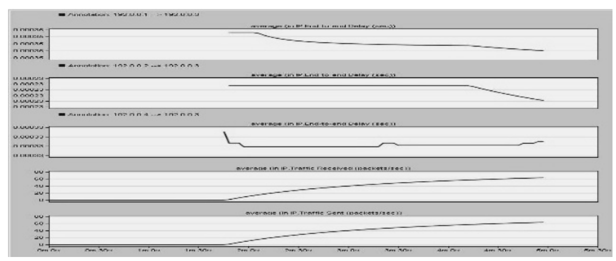

Fig. 6: Response of HTTP client

D. *The Pros and Cons of IP*

A major advantage of IP networks is that, particularly in the case of the Internet, they provide a system of global interconnectivity that is matched only by the telephone network. The IP addressing mechanism provides connectivity between any two or more end system irrespective of their position within the network hierarchy. The connectionless approach of IP has been shown to be very suitable for supporting computer data traffic, and over the years reactive congestion control methods have been developed that are extremely successful for controlling this type of traffic.

In more recent years, the emergence of distributed multimedia systems has led to the realisation that future IP networks will also need to provide some form of multiservice. A number of resource reservation protocols have been deployed that are suitable for use within IP networks. The best known of these is RSVP (Zang, *et al.*, 1993) which is a receiver oriented protocol that is optimised for multicast communication. The

RSVP model attempts to provide resource reservation whilst maintaining the connectionless principles of IP. This compromise generally means that RSVP can only provide a soft guarantee. Other resource reservation scheme, for instance, Tenet (Ferrari, *et al.*, 1994) proposes a connection oriented approach, and in these cases a firm guarantee can be given.

A resource reservation protocol is just a vehicle for requesting resources; it does not carry out the actual resource allocation. To complement RSVP, and other similar protocols, link sharing schemes (Floyd, *et al.*, 1995), queuing disciplines (Golestine, *et al.*, 2005) and scheduling algorithms (Ferrari, *et al.*, 1994) have been developed that can provide resource allocation in packet switched networks. A number of these scheduling algorithms can provide a mathematically provable delay bound, and have an associated admissions test.

In order to provide a delay bound at the packet switching level these algorithms usually require a bound on transmission delay across each individual link. Where a fixed capacity link is used, this does not present a problem. However, where the link is provided by a shared medium sub-network an addition access delay may be encountered. Methods have been developed to predict link level delay bounds in such cases (Ball, *et al.*, 1996).

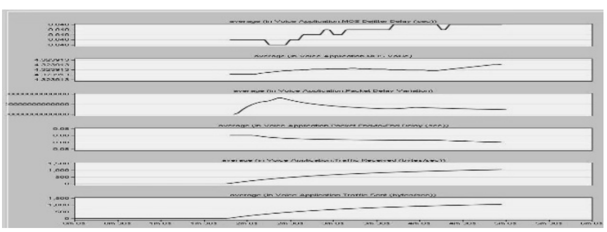

Fig. 7: Simulation of voice destination

With the inclusion of resource reservation protocols and resource allocation mechanisms IP networks should be able to add a synchronous service to the existing best effort datagram service. What remains to be shown is how closely delay can be bounded given a variable sized packet. The developers of the next generation of IP i.e. Ipv6, have taken into account the need for changes to the current Internet. Not only have they included a larger address space, but also the means to identify and control individual packet flows. It remains to be seen if the manufactures of future IP routers will take advantage of these features, or generally ignore them as they did past with the type of service information provided by IPv4.

Fig. 8: Simulation response over IP

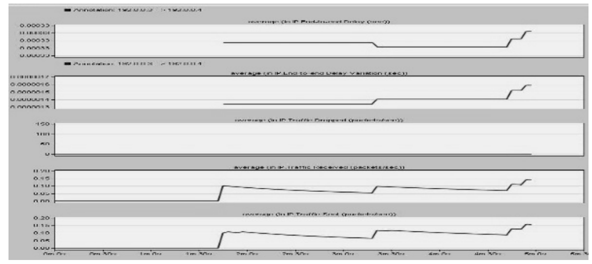

Fig. 9: Response from HTTP server

5. VOICE OVER ATM

Being a technology that can perform services in multiples, the asynchronous transfer mode has the capability of simultaneous transportation of voice, data, graphics and video at extremely high speeds.

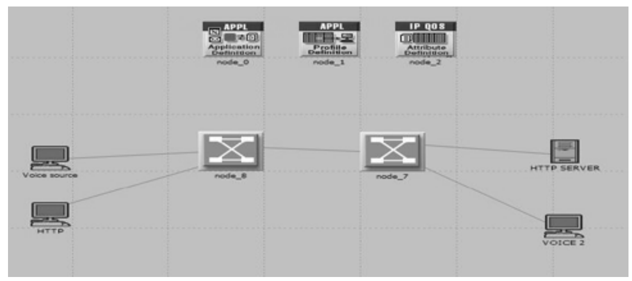

Fig.10: Topology using ATM

A. *ATM Prioritization*

By using QoS specifications, we can make good use of ATM prioritization. The CBR class was supposed to provide the best quality for voice transmission using the AAL1, but this could not happen. CBR could provide Circuit Emulation Service CES, which had the function of reducing the amount of delay. This brought even more issues. It could not wait for a little time to send a data frame; instead it is sent empty cells of fixed sizes. So actually it was not utilizing 20 bytes of bandwidth per ATM cell. On the contrary, there is an extension of the CES: Dynamic Bandwidth Circuit Emulation Service (DBCES) that can transmit only when a voice call is functioning. But as in CES, the cells might remain partially empty. Hence, the conclusion would be that AAL1 was responsible for wasting bandwidth and increasing overhead delays for VoATM.

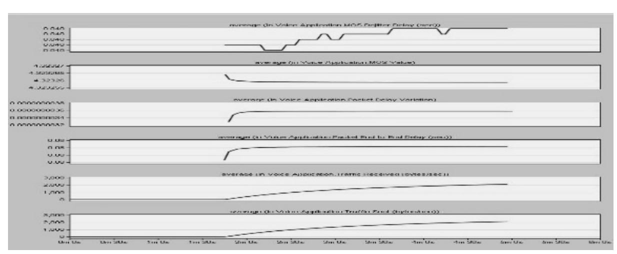

Fig. 11: Response from voice source

In order to rectify AAL1's shortcomings, AAL2 was introduced. The AAL2 was capable of providing Variable Bit Rate (VBR-RT) service that could create mini cells for ATM packets. Whichever the emulation, whether structured or not, hence, it improved efficiency of bandwidth. There were three more positive attributes of AAL2: A single ATM connection could accommodate multiple channels of voice and could exclusively support voice compression and silence suppression.

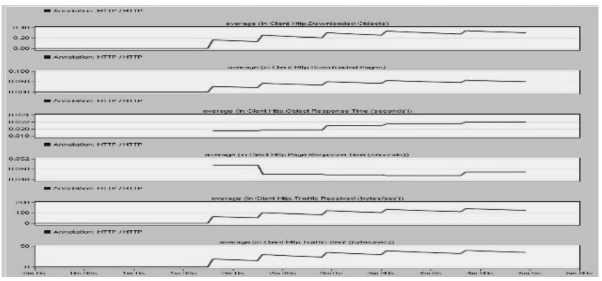

Fig. 12: Response from HTTP source

B. *The Pros and Cons of ATM*

ATM was originally proposed for use as the transport mechanism of the B-ISDN and was developed as an ITU standard (De Prycker, et al 2000). Right from the beginning it was intended that ATM should support a wide range of traffic types, including voice and video. Because of this, a major requirement of ATM was that it should combine the guarantees of circuit switching with the flexibility of packet switching.

The choice of a small fixed length cell was largely motivated by the need to share the bandwidth of the links at a fine granularity. This is clearly an advantage when multiplexing video and audio stream but is less convenient if only computer data is being supported. Furthermore, in cases where only computer data is being supported the overhead of segmentation and reassemble would seem to be an unnecessary overhead.

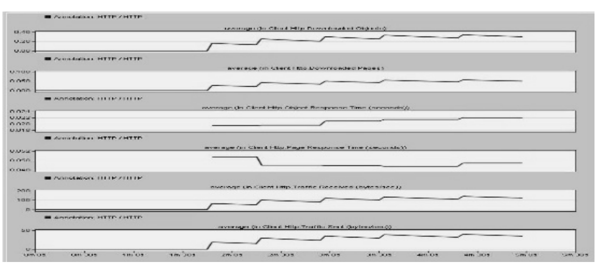

Fig.13: Simulation result of voice destination

The fact that the principles of ATM network were developed through the deliberations of a standards organisation is somewhat of a mixed blessing. On one hand, inclusion of predictive congestion control mechanisms within the standards emphasises their importance, and makes it difficult for equipment manufacturers of ATM equipment to exclude them, irrespective of implementation problems.

On the other hand, much of the research into ATM has been carried out since the standards were fixed. For example, it has never really been shown that 53 octets is an optimal size for the ATM cell.

A major criticism of ATM often cited by its critics is that it is connection oriented. Whilst this may be inconvenient for computer data communication and when interconnecting legacy LANs, it is much less of a problem and maybe and advantage, for continuous media communication. In cases where continuous media requires a guaranteed level of service the need for resource allocation implies some degree of connection. It should also be noted that methods have been developed that can provided connectionless services over ATM networks.

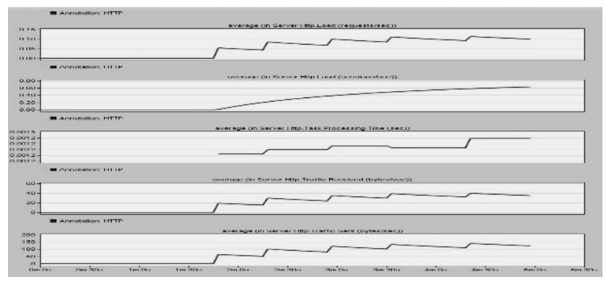

Fig. 14: Simulation result of HTTP server

It has also been demonstrated through the implementation of ATM switches that provided traffic control mechanisms are properly implemented, a fairly low delay bound can be provided. Therefore, although ATM networks cannot directly provide an isochronous service, in case where the delay bound is small, an isochronous service can be emulated without too great an increase in latency.

6. <u>ISSUES AND SOLUTIONS FOR QOS OF VOICE</u>

There are several issues of QoS that are to be looked at, and obtain a suitable solution for each of them. Echo is the consequence of getting a signal reflected back to its source. Echo is caused by delay in return time of the signal, which if it is more than 50ms causes refectory characteristics in the signal that generates an echo. The solution to this issue can be reducing all the delays between end-systems, such as accumulation delay, processing delay, and network delay. Moreover, if a digital filter is installed on the transmit path into the packet network then that could help as well.

There is another similar issue which is called jitter, a measure of the time between two packets that are passing a link. In order for this problem to be solved, the jitter buffer has to be regulated in a way that it is kept to a minimum or otherwise allowed to adjust at an allowable late packet distribution.

Then there is a crucial complication regarding the jeopardy of packets that are lost during negotiation. The nature of IP causes more probability of packets getting lost as the service is not reliable, as compared to ATM. Now in case that the packets are dropped, a repeat transmission of that information could be a nice way to deal with the matter. The redundancy will obviously avoid the voice packets from getting incomprehensive, but this would be done with further usage of bandwidth, decreasing overall efficiency.

7. CONCLUSION

QoS is necessary in most networks to differentiate between different kinds of traffic. A few parameters can be improved by simply increasing the available capacity of the links, but is often very costly. In every simulation, traffic with QoS has performed very well within its category. Traffic with "background"-quality may have high delays and low throughput, but those parameters are obviously not important for that traffic and it makes room for significant improvements for other kinds of traffic.

The downside is the needed increase in complexity, both in the network's infrastructure, such as routers, and in assigning QoS for different applications while still avoiding greedy users from trying to increase all their traffic to the highest QoS-level, thereby ruining the entire concept.

Guerin states that "Both ATM and IP each solve one half of the problem", this would seem to be true since ATM networks can certainly meet the QoS requirements of continuous media communication, whilst IP networks have demonstrated their usefulness in support computer data traffic. The argument would seem to rest with which of these two networks can most efficiently integrate the opposite type of traffic to that which it can best support.

One possible outcome of such an investigation could be to find that neither of the two networks can offer the best solution in all possible cases. Although such a result would be inconclusive, it would still prove useful for deciding the best type of network to deploy under particular circumstances.

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